



## Configuration Guide

### Configuring Transcoding in AOS

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This configuration guide outlines the use and configuration of the transcoding feature in ADTRAN Operating System (AOS) products. The guide includes an overview of the transcoding process, the steps necessary to configure transcoding using the command line interface (CLI), and transcoding troubleshooting information.

This guide consists of the following sections:

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## Transcoding Overview

Transcoding is a method of translating media types, such as between RFC 2833 events and in-band dualtone multifrequency (DTMF) signals or different voice coder-decoders (CODECs). SIP transmissions take place in the Voice over IP (VoIP) network, and include an SDP exchange over a device that links the offer and answer requests from the SIP endpoints attempting to connect a call. In AOS, the device that passes the offer and answer requests from the SIP endpoints is the AOS device acting as a session border controller (SBC). In transcoding, the SBC operates as a translator for SIP devices that are attempting a connection, but are using different media types, by translating one media type to another to allow the communication to succeed. In addition, the transcoding feature allows the SBC to filter out certain media (video) from SDP messaging. The place of the AOS SBC device in the VoIP network, and the operation of the transcoding feature, are described in *Figure 1*.



**Figure 1. AOS SBC and Transcoding in the VoIP Network**

## Transcoding Operation

When a SIP call is initiated, an SDP offer is made to the answering SIP device. This offer is passed through the SBC and along to the answering SIP device. The answering SIP device then responds with an SDP answer, which is in turn passed through the SBC to the initial offering SIP device. If there is a discrepancy between the media types (DTMF or CODEC) used by either SIP device, the SBC recognizes the mismatch and attempts to transcode the media type so that communication between the two SIP endpoints can occur.

In order to transcode the media types between the SIP endpoints, the SBC must be in the media path. It anchors both media streams between the SIP endpoints, and begins the transcoding process after receiving the media capabilities of the endpoints in the SDP offer and answer. For the transcoding process to work, the SBC must be configured with media anchoring, a feature of AOS products used to force Realtime Transfer Protocol (RTP) packets to ingress and egress the local network through the AOS device, thus anchoring the RTP stream to interfaces on the AOS device. For more information about media anchoring, and its configuration, refer to the configuration guide *Configuring Media Anchoring in AOS*, available online at <https://supportforums.adtran.com>.

There are two types of transcoding supported by the AOS SBC: DTMF interworking and CODEC transcoding. DTMF interworking is configured on a global level within the AOS device, as well as on each SIP endpoint that connects to the AOS device. When a DTMF relay mismatch occurs between an SDP offer and corresponding answer, the SBC transcodes between the two formats to allow the call to be completed. In-band and RFC 2833 named telephone events (NTE) are the only DTMF relay methods that are acceptable as a transcoding source or target. In addition, if an answering SIP endpoint is configured for NTE with a payload type of *X*, and an offer is destined for the endpoint with an NTE payload type *Y*, transcoding is not used by the SBC, but rather, the SBC sends the offer with payload type *Y* to the answering endpoint. This method saves transcoding resources on the SBC.

CODEC transcoding is also configured on a global level within the AOS device, as well as on each SIP endpoint that connects to the AOS device. CODEC lists can be configured on SIP endpoints, which affect the filtering of CODECs in both the transmit and receive directions, or to a specific traffic direction. For example, you can allow any CODEC to flow into the SBC and restrict transmissions to a single CODEC, transcoding as necessary. In addition, you can specify that any CODEC is allowed to flow to or from the SIP endpoint.

Any CODECs that are added in the translated SDP offer are appended to the end of the CODEC list to prevent transcoding when it is unnecessary (if the answering device chooses a higher priority non-transcoded CODEC first). If the SBC device receives an offer that does not contain any compatible source CODECs, then transcoding is not possible. *Table 1* describes the source and target CODECs supported by AOS devices. Note that transcoding between the varieties of G.729 is not supported; rather, if any G.729 CODEC is offered, it can pass through but is not transcribed. In addition, note that transcoding to or from T.38 is not supported, but that companding type conversion is supported (G.711u to G.711a and vice-versa).

**Table 1. Supported Source and Target CODECs**

Source CODECs	Target CODECs
G.711a	G.711a
G.711u	G.711u
G.729	G.729a G.729ab
G.729a	
G.729b	
G.729ab	

Whether DTMF or CODEC transcoding is being employed, a digital signal processor (DSP) is required. Software transcoding for devices without a DSP is not supported. Because transcoding uses the DSP, every call that requires transcoding consumes exactly two DSP channels. These channels do not have to be contiguous, but they must both reside on the same DSP; for example, a single channel on DSP 1 and a single channel on DSP 2 is not sufficient to transcode. The two channel requirement is true regardless of the source and target CODECs or the DTMF relay methods used, but two channels are sufficient for transcoding both CODEC and DTMF relays simultaneously on the same call.

*Table 2* outlines the supported CODECs, DTMF relay methods, transcoding functions, and the required DSP channels for each transcoding operation.



*In-band DTMF signaling over G.729 is supported, but it is not recommended.*

**Table 2. Supported Transcoding Scenarios**

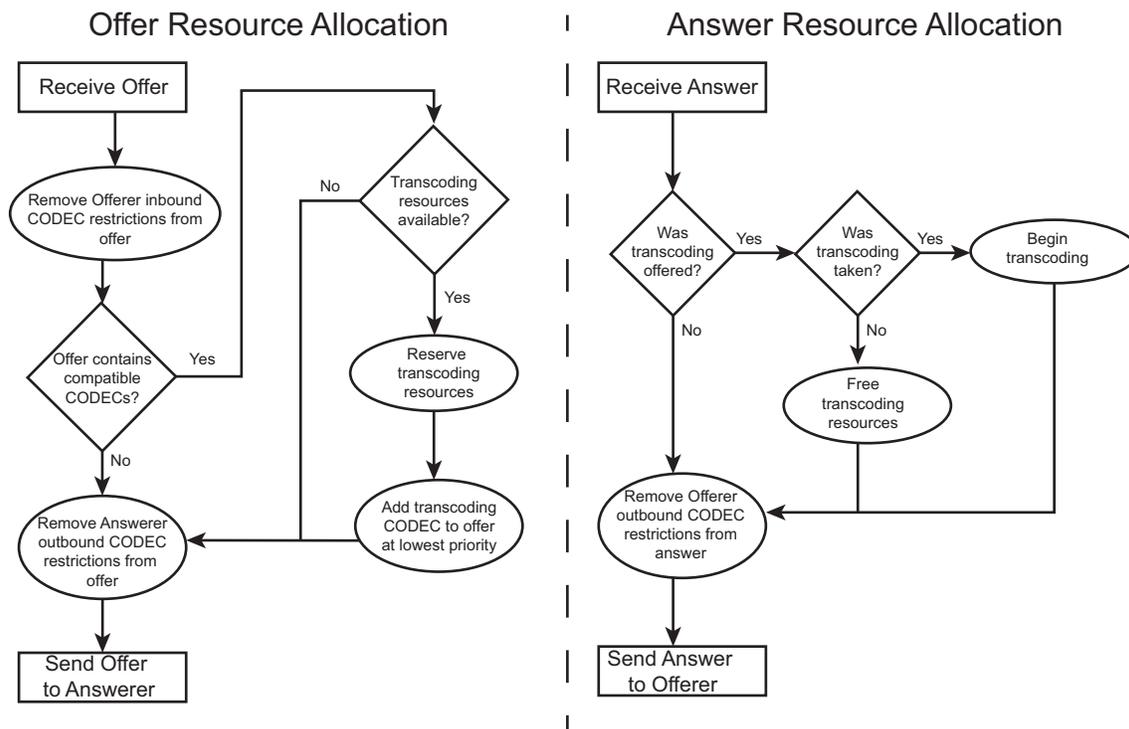
Offer CODEC	Offer DTMF	Answer CODEC	Answer DTMF	Transcode CODEC	Transcode DTMF	Required DSP Channels	
g711u	In-Band (IB)	g711u	IB	No	No	0	
			NTE	No	Yes	2	
		g711a	IB	Yes	No	2	
			NTE	Yes	Yes	2	
		g729a	IB	Yes	No	2	
			NTE	Yes	Yes	2	
	NTE	g711u	IB	No	Yes	2	
			NTE	No	No	0	
		g711a	IB	Yes	Yes	2	
			NTE	Yes	No	2	
		g729a	IB	Yes	Yes	2	
			NTE	Yes	No	2	
	g711a	IB	g711u	IB	Yes	No	2
				NTE	Yes	Yes	2
g711a			IB	No	No	0	
			NTE	No	Yes	2	
g729a			IB	Yes	No	2	
			NTE	Yes	Yes	2	
NTE		g711u	IB	Yes	Yes	2	
			NTE	Yes	No	2	
		g711a	IB	No	Yes	2	
			NTE	No	No	0	
		g729a	IB	Yes	Yes	2	
			NTE	Yes	No	2	

**Table 2. Supported Transcoding Scenarios (Continued)**

Offer CODEC	Offer DTMF	Answer CODEC	Answer DTMF	Transcode CODEC	Transcode DTMF	Required DSP Channels
g729 (any)	IB	g711u	IB	Yes	No	2
			NTE	Yes	Yes	2
		g711a	IB	Yes	No	2
			NTE	Yes	Yes	2
		g729 (same)	IB	No	No	0
			NTE	No	Yes	2
	NTE	g711u	IB	Yes	Yes	2
			NTE	Yes	No	2
		g711a	IB	Yes	Yes	2
			NTE	Yes	No	2
		g729 (same)	IB	No	Yes	2
			NTE	No	No	0

## Transcoding Resource Considerations

When the SBC receives an SDP offer, it must decide if transcoding is necessary or not, regardless of whether the offer comes from the session originator or the session destination. If the offer contains compatible CODECs based on the offerer's inbound CODEC restrictions, then transcoding is an option. If the offer does not already contain all of the supported target CODECs (see [Table 1 on page 3](#)), then the SBC can add additional CODECs to the offer as long as the answering device's outbound CODEC restrictions allow it. If at this point, a transcoding opportunity exists, the SBC attempts to reserve transcoding resources. If resources are available, the SBC appends available transcoding CODECs to the offer and passes it along. If resources are not available, the SBC does not add any additional CODECs to the offer. This allows the call to complete, assuming there are CODEC compatibilities between the offering and answering devices. The transcoding resources for the call are reserved at the reception of the SDP offer. If the answering device has chosen a CODEC that was not present in the original offer, that means it has selected a CODEC that was added to the offer by the SBC, and transcoding takes place. If transcoding is either unnecessary or not possible, the resources allocated for the call are freed and can be used by other parts of the system. If transcoding does take place, the resources will not be freed for the duration of the call, even if the session media is altered in such a way that transcoding is no longer necessary. [Figure 2 on page 6](#) outlines the transcoding resource allocation process in the AOS SBC.



**Figure 2. SBC Transcoding Resource Allocation**

Resource allocation for transcoding purposes is done dynamically following the process outlined above. In addition, if a SIP INVITE message is received by the SBC that does NOT contain an SDP offer, transcoding resources are not reserved. The SBC only attempts to reserve transcoding resources when an SDP offer is received. Once the SBC receives the SDP offer, it must determine the destination SIP endpoint (answering endpoint). If the offer and the answering device's outbound CODEC restrictions do not match, meaning that transcoding is necessary, then transcoding resources are required. If these resources are not available, the call is rejected with a 503 (Service Unavailable) error. Should this event occur, a warning event is created in the event log. If this event occurs when transcoding is a possibility, but not absolutely required, and insufficient resources are available, a debug message is logged instead.



*A 503 error occurs only if the transcoding resources are not available AND transcoding is enabled. If transcoding is not enabled, and the offer and the answering device's outbound CODEC restrictions do not match, a 415 (Unsupported Media) error occurs.*

## Call Throughput and Call Delay Considerations

In addition to using SBC resources, transcoding can affect the call throughput and the call delay in the SBC network. Keep in mind the following considerations to properly implement transcoding on your network with minimal undesired effects.

The maximum call throughput for transcoded calls is directly proportional to the number of DSP channels available on your AOS SBC. Every transcoded call requires two DSP channels, which limits the maximum call volume on the SBC device to half of the number of DSP channels if every call is transcoded. In addition, because transcoding requires media anchoring to be enabled, and many network configurations depend on voice quality monitoring being enabled, the overhead of both features causes the SBC call capacity to be reduced to 60 percent of normal. This penalty is incurred as long as transcoding is enabled, even if no calls are transcoded because 60 percent is the upper bound for an AOS SBC device with media anchoring enabled (regardless of whether transcoding is enabled).

In addition, transcoding adds a minimum call path delay of 60 ms. This delay is the same for all transcoded streams, regardless of the CODEC or DTMF relay mode used.

## Hardware and Software Requirements and Limitations

In addition to the resource and network considerations discussed in *Transcoding Overview on page 2*, the transcoding feature is only supported on specific AOS SBC platforms. For a complete list of supported platforms, refer to the *AOS Feature Matrix*, available online at <https://supportforums.adtran.com>.

The number of supported simultaneous transcoding calls on a specific platform will depend on the number of DSP channels available on that platform.

Software transcoding for devices without a DSP is not supported in AOS firmware release R10.4.0.

Media anchoring must be enabled for transcoding to function properly. For more information about media anchoring, and its configuration, refer to the configuration guide *Configuring Media Anchoring in AOS*, available online at <https://supportforums.adtran.com>.

The G.711 CODEC is the only recommended CODEC for use with in-band DTMF signaling, but other lower bit-rate CODECs (for example, G.729) are supported.

When DTMF interworking is configured on a SIP endpoint, it does not restrict DTMF relay methods for the endpoint. For example, an endpoint configured for NTE can still send offers and answers to the SBC using in-band DTMF; however, the SBC will only send NTE offers to the endpoint. In addition, transcoding does not occur between different NTE payload types.

Because you can configure a CODEC list to support any CODEC (using the **any** keyword), the string *any* cannot be used as a CODEC list name.

Transcoding of T.38 is not supported. In addition, transcoding between the varieties of G.729 is not supported, although if any G.729 CODEC is offered, it can be passed through to the answering endpoint.

Transcoding is not supported with the SIP proxy or asymmetric audio media streams.

## Configuring DTMF Transcoding Using the CLI

To configure DTMF transcoding, perform the following tasks:

1. Access the CLI.
2. Enable media anchoring.
3. Enable DTMF transcoding globally on the AOS SBC device.
4. Configure DTMF relay settings for SIP endpoints.

## Step 1: Accessing the CLI

To access the CLI on your AOS unit, follow these steps:

1. Boot up the unit.
2. Telnet to the unit (**telnet** <ip address>), for example:

**telnet 10.10.10.1.**



*If during the unit's setup process you have changed the default IP address (10.10.10.1), use the configured IP address.*

3. Enter your user name and password at the prompt.



*The AOS default user name is **admin** and the default password is **password**. If your product no longer has the default user name and password, contact your system administrator for the appropriate user name and password.*

4. Enable your unit by entering **enable** at the prompt as follows:

**>enable**

5. If configured, enter your Enable mode password at the prompt.

6. Enter the unit's Global Configuration mode as follows:

**#configure terminal**  
(config)#

## Step 2: Enabling Media Anchoring

Enable media anchoring using the **ip rtp media-anchoring** command from the Global Configuration mode. This command enables media anchoring on a global basis for all RTP-to-RTP calls. By default, media anchoring is disabled. Using the **no** form of this command disables the media anchoring feature.

To enable media anchoring, enter the command as follows from the Global Configuration mode:

```
(config)#ip rtp media-anchoring  
(config)#
```

At this point in the configuration, you can optionally choose to configure other media anchoring settings. Refer to the configuration guide *Configuring Media Anchoring in AOS* available online at <https://supportforums.adtran.com> for more information.

## Step 3: Enable DTMF Transcoding Globally

Enable DTMF transcoding globally on the AOS SBC device using the **ip rtp media-anchoring transcoding dtmf** command from the Global Configuration mode. This command enables DTMF transcoding on a global basis for all SIP endpoints. By default, DTMF transcoding is disabled. Using the **no** form of this command disables the DTMF transcoding feature.

To enable DTMF transcoding, enter the command as follows from the Global Configuration mode:

```
(config)#ip rtp media-anchoring transcoding dtmf
(config)#
```

#### Step 4: Configure DTMF Transcoding Settings for SIP Endpoints

After enabling DTMF transcoding globally on the AOS SBC device, you can optionally specify which DTMF relay mode to offer a particular endpoint. Specify the DTMF mode using the **rtp dtmf-relay offer [inband | nte <value>]** command from the SIP trunk or voice user's configuration mode. Valid **nte** range is **96** to **127**. By default, there is no preferred DTMF relay mode, and the SBC can choose a DTMF value that does not require transcoding to complete a call. Using the **no** form of this command returns the DTMF method to the default value. Configure this setting for each SIP endpoint (SIP trunk) that will be using transcoding.

To specify the DTMF relay mode used for transcoding, enter the command as follows from the SIP Trunk Configuration mode as follows:

```
(config)#voice trunk T01 type sip
(config-T01)#rtp dtmf-relay offer nte 101
(config-T01)#
```

After you have configured the DTMF relay modes for each SIP endpoint that will use transcoding, the configuration for DTMF transcoding is complete.

## Configuring CODEC Transcoding Using the CLI

To configure CODEC transcoding, perform the following tasks:

1. Access the CLI.
2. Enable media anchoring.
3. Enable CODEC transcoding globally on the AOS SBC device.
4. Configure CODEC settings for SIP endpoints.

### Step 1: Accessing the CLI

To access the CLI on your AOS unit, follow these steps:

1. Boot up the unit.
2. Telnet to the unit (**telnet <ip address>**), for example:

```
telnet 10.10.10.1.
```



*If during the unit's setup process you have changed the default IP address (10.10.10.1), use the configured IP address.*

3. Enter your user name and password at the prompt.



*The AOS default user name is **admin** and the default password is **password**. If your product no longer has the default user name and password, contact your system administrator for the appropriate user name and password.*

4. Enable your unit by entering **enable** at the prompt as follows:

```
>enable
```

5. If configured, enter your Enable mode password at the prompt.

6. Enter the unit's Global Configuration mode as follows:

```
#configure terminal  
(config)#
```

## Step 2: Enabling Media Anchoring

Enable media anchoring using the **ip rtp media-anchoring** command from the Global Configuration mode. This command enables media anchoring on a global basis for all RTP-to-RTP calls. By default, media anchoring is disabled. Using the **no** form of this command disables the media anchoring feature.

To enable media anchoring, enter the command as follows from the Global Configuration mode:

```
(config)#ip rtp media-anchoring  
(config)#
```

At this point in the configuration, you can optionally choose to configure other media anchoring settings. Refer to the configuration guide *Configuring Media Anchoring in AOS* available online at <https://supportforums.adtran.com> for more information.

## Step 3: Enable CODEC Transcoding Globally

Enable CODEC transcoding globally on the AOS SBC device using the **ip rtp media-anchoring transcoding codec** command from the Global Configuration mode. This command enables CODEC transcoding on a global basis for all SIP endpoints. By default, CODEC transcoding is disabled. Using the **no** form of this command disables the CODEC transcoding feature.

To enable CODEC transcoding, enter the command as follows from the Global Configuration mode:

```
(config)#ip rtp media-anchoring transcoding codec  
(config)#
```

## Step 4: Configure CODEC Settings for SIP Endpoints

After enabling CODEC transcoding globally on the AOS SBC device, you can optionally specify the CODEC settings for a particular endpoint by creating a CODEC list and applying the list to the trunk or user.

To create a CODEC list, and apply it to a trunk or user, follow these steps:

1. Create a CODEC list using the **voice codec-list <name>** command from the Global Configuration mode. The **<name>** parameter is the name of the list that you will then apply to voice trunks and voice users. Using this command enters the CODEC List Configuration mode. Enter the command as follows:

```
(config)#voice codec-list G711_ONLY
(config-codec)#
```



*Because you can choose to specify that any CODEC is used by a SIP endpoint with the **any** keyword, you cannot create a CODEC list with a name of **any**.*

2. From the CODEC List Configuration mode, specify the CODECs, in the order of preference, that will be used by the voice trunk or user by entering the **codec [g711alaw | g711ulaw | g722 | g729]** command. The **g711alaw** parameter specifies that the G.711 A-Law CODEC is used (64000 bps), the **g711ulaw** parameter specifies that the G.711 mu-Law CODEC is used (64000 bps), the **g722** parameter specifies that the G.722 CODEC is used, and the **g729** parameter specifies that the G.729 CODEC is used (8000 bps). Enter the command multiple times to specify the order of preference for the CODECs in the list. When calls are negotiated, the voice trunk or user interface will use these CODECs in the order they are listed. Use the **no** form of this command to remove a CODEC from the CODEC list. To specify that the CODEC list **G711\_ONLY** uses G.711 mu-Law first, and then G.711 A-Law, enter the command as follows:

```
(config-codec)#codec g711ulaw
(config-codec)#codec g711alaw
```



*Although you can specify a voice trunk or user to use the G.722 CODEC, it is not supported by CODEC transcoding.*

3. After configuring the CODEC list, apply the list to the voice trunk(s) or user(s) that will be using transcoding by entering the **codec-list <name> [both | in | out]** or **codec-list any** commands from the Voice Trunk or Voice User Configuration modes. You can apply a previously created CODEC list to the SIP endpoint using the **codec-list <name>** command, and then can specify the traffic direction to which the CODEC applies by optionally specifying **both** (transmitted and received SDP), **in** (received SDP only), or **out** (transmitted SDP only). If no direction is specified, the CODEC list is applied to SDP in both directions. You can also specify that the SIP endpoint allows any possible CODEC by entering the **codec-list any** command. To specify that the CODEC list **G711\_ONLY** is applied to incoming SDP traffic on a voice trunk or voice user, enter the command as follows:

```
(config)#voice trunk T01
(config-T01)#codec-list G711_ONLY in
```

OR

```
(config)#voice user 2004
(config-2004)#codec-list G711_ONLY in
```

To specify that any possible CODEC is accepted by the voice trunk or user, enter the command as follows:

```
(config)#voice trunk T01  
(config-T01)#codec-list any
```

OR

```
(config)#voice user 2004  
(config-2004)#codec-list any
```



*Because you can choose to specify that any CODEC is used by a SIP endpoint with the **any** keyword, you cannot create a CODEC list with a name of **any**.*

After you have configured the CODEC settings for each SIP endpoint that will use transcoding, the configuration for CODEC transcoding is complete.

## Configuring Media Filtering Using the CLI

In addition to DTMF and CODEC transcoding, the transcoding feature also allows you to filter out video media attributes from transmitted or received SDP. By default, this feature is disabled and all video SDP media attributes are passed through the SBC device. You can filter out video media on a SIP endpoint by entering the **rtp media video filter** command from the SIP endpoint's configuration mode. Using the **no** form of this command disables the feature.

To enable video media filtering on a voice trunk, enter the command from the trunk's configuration mode as follows:

```
(config)#voice trunk T01  
(config-T01)#rtp media video filter  
(config-T01)#
```

## Transcoding Configuration Examples

The example scenarios contained within this section are designed to enhance understanding of transcoding configurations on AOS products. The examples describe some of the common real-world transcoding configurations. All configurations provided in this section use the CLI.



*The configuration parameters entered in these examples are sample configurations only. These applications should be configured in a manner consistent with the needs of your particular network. CLI prompts have been removed from the configuration examples to provide a method of copying and pasting configurations directly from this configuration guide into the CLI. These configurations should not be copied without first making the necessary adjustments to ensure they will function properly in your network.*

In the following example, DTMF relay transcoding is configured on the AOS SBC unit, with a SIP trunk configured to use DTMF relay mode **nte 101**.

```
!
ip rtp media-anchoring
ip rtp media-anchoring transcoding dtmf
!
voice trunk T01 type sip
  rtp dtmf-relay offer nte 101
!
!
```

In the following example, CODEC transcoding is configured on the SBC device using the CODEC list **G711\_ONLY**, which uses the G.711 mu-law and the G.711 A-law CODECs for SDP in both directions on SIP trunk **T01**, with video filtered out of the SDP transmissions.

```
!
ip rtp media-anchoring
ip rtp media-anchoring transcoding codec
!
voice codec-list G711_ONLY
  codec g711ulaw
  codec g711alaw
!
!
voice trunk T01 type sip
  codec-list G711_ONLY both
  rtp media video filter
!
```

## Transcoding Configuration Command Summary

The following table summarizes the commands used to configure transcoding in AOS products.

**Table 3. Transcoding Configuration Command Summary**

Prompt	Command	Description
(config)#	<b>[no] ip rtp media-anchoring</b>	Enables or disables media anchoring. Media anchoring must be enabled for transcoding to function. Media anchoring is disabled by default.
(config)#	<b>[no] ip rtp media-anchoring transcoding dtmf</b>	Globally enables or disables DTMF transcoding. DTMF transcoding is disabled by default.

**Table 3. Transcoding Configuration Command Summary (Continued)**

Prompt	Command	Description
(config-T01)#	<b>[no] rtp dtmf-relay offer [inband   nte &lt;value&gt;]</b>	Specifies the DTMF transcoding settings for a SIP endpoint. The <b>inband</b> parameter specifies that the DTMF relay method is in-band, and the <b>nte</b> parameter specifies that the DTMF relay method is NTE. NTE value range is <b>96</b> to <b>127</b> . By default, there is no preferred DTMF relay mode on the SIP endpoint, and the SBC can choose a DTMF value that does not require transcoding to complete a call.
(config)#	<b>[no] ip rtp media-anchoring transcoding codec</b>	Globally enables or disables CODEC transcoding. CODEC transcoding is disabled by default.
(config)#	<b>[no] voice codec-list &lt;name&gt;</b>	Creates a CODEC list to apply to a SIP endpoint.
(config-codec)#	<b>[no] codec [g711alaw   g711ulaw   g722   g729]</b>	Specifies the order of preference and the CODECs used in a CODEC list.
(config-T01)#	<b>[no] codec-list &lt;name&gt; [both   in   out]</b>	Applies a CODEC list to a SIP endpoint, and specifies to which SDP traffic direction the CODEC list is applied. If no direction is specified, it is applied to both received and transmitted SDP.
(config-T01)#	<b>[no] codec-list any</b>	Specifies that the SIP endpoint accepts any possible CODEC.
(config-T01)#	<b>[no] rtp media video filter</b>	Enables or disables the filtering of video media from received and transmitted SDP. By default, video media filtering is disabled.

## Troubleshooting

The following **show** commands can be used to view RTP resources and their allocation in the SBC device, or to display all of the anchored RTP flow associations and the number of relayed packets per association currently active in an anchored RTP flow. These commands are all entered from Enable mode.

Use the **show ip rtp media sessions** command to display all of the anchored RTP flow associations and the number of relayed packets per association currently active in an anchored RTP flow. In addition, the TTL and the session type (DSP, media-anchored, or transcoded) for the association is displayed.

To display media information, enter the command as follows:

>enable

#show ip rtp media sessions

Call	Anchored Address	Remote Address	TTL	Pkts	Ovrd	Type	Sess
6	10.10.10.1:10020	10.10.10.2:2232	45	506	No	Audio	Xcode
6	10.17.250.8:10022	10.17.250.5:50000	45	509	No	Audio	Xcode
6	10.10.10.1:10021	10.10.10.2:2233	41	2	No	Audio	Xcode
6	10.17.250.8:10023	10.17.250.5:50001	45	2	No	Audio	Xcode

Use the **show rtp resources** command to display the status of the DSP resources on the AOS unit. Enter the command as follows:

>enable

#show rtp resources

DSP	Channel	Type	Interface	Port	Status
0/1	1	RTP	N/A	0	Available
0/1	2	RTP	N/A	0	Available
0/1	3	RTP	N/A	0	Available
0/1	4	RTP	N/A	0	Available
0/1	5	RTP	N/A	0	Available
0/1	6	RTP	N/A	0	Available
0/1	7	RTP	N/A	0	Available
0/1	8	RTP	N/A	0	Available
0/1	9	RTP	N/A	0	Available
0/1	10	RTP	N/A	0	Available
0/1	11	RTP	N/A	0	Available
0/1	12	RTP	N/A	0	Available
0/1	13	RTP	N/A	0	Available
0/1	14	RTP	N/A	0	Available
0/1	15	RTP	N/A	0	Available
0/1	16	RTP	N/A	0	Available
0/1	17	RTP	N/A	0	Available
0/1	18	RTP	N/A	0	Available
0/1	19	RTP	N/A	0	Available
0/1	20	RTP	N/A	0	Available
0/1	21	RTP	N/A	0	Available
0/1	22	RTP	N/A	0	Available
0/1	23	RTP	N/A	0	Available
0/1	24	RTP	N/A	0	Available
0/1	25	RTP	N/A	0	Available
0/1	26	RTP	N/A	0	Available
0/1	27	RTP	N/A	0	Available
0/1	28	RTP	N/A	0	Available
0/1	29	RTP	N/A	0	Available
0/1	30	RTP	N/A	0	Available